REMARKS/ARGUMENTS

Claims 1-12 now stand in the present application, claims 1, 2, 4 and 7-10 having been amended and new claims 11-12 having been added. Reconsideration and favorable action is respectfully requested in view of the above amendments and the following remarks.

In the Final Office Action of August 5, 2008, the Advisory Action of December 2, 2008, and the Examiner's Answer of May 19, 2009, the Examiner has rejected claims 1-10 under 35 U.S.C. § 103 as being unpatentable over Aharoni in view of Zhu. In view of the above-described claim amendments the Examiner's § 103 rejection of the claims is believed to have been overcome.

Applicants have amended the claims to incorporate the terminology of "maximum timing error," which the Examiner had previously objected to as not adequately defined in the claim. The amended claims are believed to overcome the prior indefiniteness rejection and are believed to be fully supported by the present specification as explained below.

It will be helpful to discuss the concept of timing error in the context of the present specification at page 4 which describes a transmitter which transmits a video sequence of N packets, packet 0 to packet N, sequentially one at a time to a receiver. *See*, specifically, present specification at paragraph 1 of page 4. At a certain time during the transmission of the sequence, the receiver starts to queue received packets in a receiving buffer prior to decoding. Consider the situation where packets 1 to g-1 have been decoded, g to h-1 are held in the buffer for decoding, and the remaining packets h to N have yet to be received by the receiver. *See,* present specification at

page 4, lines 13 to 15. The amount of data which has yet to be sent (h-N) is effectively a sub-sequence of the video sequence (packets 0-N) whose transmission has already begun and which is being decoded.

The time available for the transmission duration of a packet j is the time from the point at which a packet j is to be transmitted until the time it needs to have been received which is the time the previous packet has finished being decoding and at which the receiving buffer would otherwise become empty. *See,* present specification at page 4, lines 16 to 19. Now, the total amount of data bits which have yet to be received by the receiver is thus given by the sum of the bits of each packet left in the sequence. This means summing the bits of each packet for packets h to j, so that the data left to send can be expressed as $\sum_{i=h}^{i=j} b_i$. *See*, present specification at Equation 2 on page 4. Now, at any particular point in time during the transmission of all of these data bits, the bits must be received by the receiver before the receiving buffer empties. So, each packet j must be received in time to be decoded, which will be satisfied if the condition set out in Equation 4 in the description is met, namely that $\frac{\sum_{i=h}^{i=j} b_i}{R} \leq t_j - t_g$.

The above equation considers a series of packets which is transmitted sequentially from packet h to packet j. As the number of bits in each packet varies (i.e., b_i varies for each packet), even with the transmission rate is assumed to be constant, some packets will arrive sooner than others. If at any time a packet arrives later than the time available, provided there are packets already held in the queue when it does arrive, the error in the time of arrival of that packet can be compensated for. Thus, over

the series of sequentially transmitted packets, even though the duration of the transmission time for each packet may vary, as long as for each of the packets forming the series the receiving buffer can compensate for the variation, i.e., as long as there are a sufficient number of packets held in the receiving buffer at the time each packet is sent to compensate for their variation in transmission times, the receiver buffer will not underflow/empty.

The present specification at page 5, lines 14 and 15 indicates that the condition for packets h to j to avoid buffer underflow is given by $\sum_{i=h}^{j} \Delta \varepsilon_i = \sum_{i=h}^{j} \frac{b_i}{R} - (t_i - t_{i-1}) \leq T_B$, where T_B is the current amount of buffered information that the client has. Accordingly, the maximum timing error for a sequence of packets that may occur for the transmission of packet h up to the end of the sequence N-1 (as we have N packets in the sequence and i = 0,...,N-1) is given on page 5 by Eqn. (7), as $Max_{j=h}^{j=N-1}\left\{\sum_{i=h}^{j} \Delta \varepsilon_i\right\} \leq t_{h-1} - t_g$.

This means that the maximum error in the time at which the receiving buffer receives any of packets h to N-1 cannot exceed the ability of the receiving buffer to compensate for that error, which will be dependent on the contents of the receiving buffer at the time the delayed packet is received (and consequently also on the contents at the time the packet is sent). If one or more packets are received too soon, it could cause the receiving buffer to overflow, if one or more packets are sent too slowly, it could cause the receiving buffer to underflow. However, the ability of the receiving buffer to be able to compensate for a timing error will depend on where in the timing sequence the error occurs (as this will affect what packets are already queued in the receiving buffer queue).

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Prior Art Comments

Zhu et al teaches that if the maximum end to end delay is known, and extra delay called a smoothing delay can be added so that at the receiver the overall delay = queuing delay + smoothing delay (applied by the receiver after the packets have been transmitted before being held in receiver buffer) is constant. *See*, Zhu at Col. 1, lines 28 to 34, and Figure 1.

Zhu considers how to figure out what is the maximum end to end delay that the receiving buffer can tolerate so it doesn't overflow – i.e., what is the smallest amount of smoothing delay which can be added so that the overall delay does not result in the receiver's decoding buffer overflowing. *See*, Zhu at Col. 2, lines 54 to 56.

The maximum timing error is the maximum amount of time a packet's arrival in the receiving buffer can be delayed by if the buffer is not to underflow, whereas the maximum end-to-end delay is the maximum amount of time that the decoding of a packet can be delayed by if the receiving buffer is not to overflow. The maximum end-to-end delay in Zhu is <u>not</u> the same as the maximum timing error as disclosed and claimed in the present application.

In Zhu, at Col. 3, lines 3 to 25, especially, lines 22 to 24 describe how, in order to avoid buffer overflow or underflow at the receiver due to the variable delays that packets can encounter, the receiver's decoder buffer verifier (DBV) checks the buffer fullness at each frame interval. In order to smooth the delay but avoid buffer overflow, the DBV checks to find out what is the smallest number of stuffing bits which can be added to minimise the overall delay incurred at the decoder.

It is quite clear, therefore, that Zhu does not calculate at the transmitting end what is the variable delay a packet may encounter prior to that packet being transmitted. This is because in fact, Zhu does not care what the transmission delay is as Zhu does not consider buffer underflow a problem – see Zhu at Col. 3, lines 27 to 32 – as they describe the decoder just repeatedly displaying the same video frame while waiting for the next frame to arrive. In Zhu, they simply assume that if the worse network delay is D – then underflow can occur at most D/T times, where T is the frame period. Col. 5, lines 1 to 40 of Zhu, consider only what the level of buffer fullness will be after a frame has been removed from the decoder in the sense of avoiding buffer OVERFLOW. *See*, Zhu at Col. 5, line 4.

Aharoni et al., at Col. 17, lines 52 to 59, teaches that "*during real-time transmission of video data, the client reports back status ad bandwidth related information to the video server via a reverse channel. Based on the number of transmission errors as well as the number of data packets lost, as communicated by the status and bandwidth information sent back to the server, the server makes an online determination regarding the quantity of data to send to the client.*" The server according to Aharoni selects "*a compression level for a data stream to ensure that when transmitted to a receiver, the transmitted data fully occupies the available bandwidth of the network connection between the transmitter and a receiver.*" *See*, Aharoni et al., at Col. 8, lines 3 to 17.

Aharoni et al. teaches that the transmission of the packets over the network is controlled by a rate control unit 106 which "*keeps track of the amount of video in terms of time that is queued for display at the client*" by using "*acknowledges received by*

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client via the acknowledgement server 108 to determine the next packet transmission time." *See*, Aharoni et al., at Col.12, lines 63 to 64. Before transmission, a packet is copied and one copy of the packet is held in a buffer so the copy of the packet can be sent if an acknowledgement from the receiving client for the original packet is not received in time.

Accordingly, in Aharoni et al., a particular compression level is selected to ensure that the available bandwidth of a network connection between the transmitting server and each client is fully utilised, noting that this can vary for different clients. The timing of the transmission of the packets over the network is used to determine when to send each packet out over the connection once the rate has been selected. Nothing in Aharoni et al. teaches determining a timing error in the manner of the invention as claimed, namely, for each candidate version of a data sequence, prior to transmission, calculating the maximum timing error for a subsequence of one or more portions of that version of the video sequence.

This is advantageous as if the number of bits b_i rate of each portion is not constant it means *the timing error of a discrete portion of a candidate version which is the difference between the time needed to transmit the discrete portion at the currently ascertained permitted transmission data rate and the difference in time between the playing instant of the respective portion at the receiving buffer and the preceding playing instant of a portion received by said receiving buffer, can vary for each portion.*

It is thus non-trivial to determine the maximum value of the timing error for a subsequence of one or more data portions. As set out in amended claim 4, the timing error of a subsequence which starts with "the portion in question", is determined by the

total time to transmit at the relevant rate, the portion in question and zero or more consecutive subsequent portions up to and including any particular portion, and (b) the difference between the playing instant of the respective particular portion and the playing instant of the portion preceding the portion in question.

If bi can vary for each portion (or packet) in a data sequence, it is quite possible for some portions in a sequence to arrive late and some portions sooner than would result from the average time of transmission for the data sequence, so that, even if the transmission time of an individual portion deviates by a large amount, as long as when it is received by the receiver, at that instant the receiving buffer is ascertained to have the capability to cope with the variation in transmission duration, the currently ascertained candidate version can be selected for transmission.

Nothing in Aharoni et al. teaches or prompts a person of skill in the art to calculate the maximum timing error that a receiving buffer can tolerate for an entire version of a data sequence in the manner of the invention, i.e., on a such a data portion by data portion basis for any number of data portions (where the data sequence comprises a sequence of plural such data portions).

Accordingly, it is submitted that the invention as set out in the amended claims is clearly not anticipated by any combination of Zhu and/or Aharoni.

Therefore, in view of the above amendments and remarks, it is respectfully requested that the application be reconsidered and that all of claims 1-12, now standing in the application, be allowed and that the case be passed to issue. If there are any other issues remaining which the Examiner believes could be resolved through either a

supplemental response or an Examiner's amendment, the Examiner is respectfully requested to contact the undersigned at the local telephone exchange indicated below.

Respectfully submitted,

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